Avaya Solution & Interoperability Test Lab

Application Notes for Algo 8028 SIP Doorphone with Avaya Aura® Session Manager and Avaya Aura® Communication Manager - Issue 1.0

Abstract

These Application Notes describe the steps required to integrate the Algo 8028 SIP Doorphone with Avaya Aura® Session Manager and Avaya Aura® Communication Manager configured as an Evolution Server. The 8028 SIP Doorphone provides hands-free intercom capability and entrance security with door unlock control. It is a SIP compliant device that registers with Session Manager. The 8028 Doorphone includes a Control Unit and Door Station.

Information in these Application Notes has been obtained through DevConnect compliance testing and additional technical discussions. Testing was conducted via the DevConnect Program at the Avaya Solution and Interoperability Test Lab.

1. Introduction

These Application Notes describe the steps required to integrate the Algo 8028 SIP Doorphone with Avaya Aura® Session Manager and Avaya Aura® Communication Manager configured as an Evolution Server. The 8028 SIP Doorphone provides hands-free intercom capability and entrance security with door unlock control. It is a SIP compliant device that registers with Session Manager. The 8028 Doorphone includes a Control Unit and Door Station.

A visitor can press the call button on the Door Station to ring a specified telephone. The called party can then answer the call to communicate with the Door Station. Using DTMF tones, the called party can press a digit on the phone keypad to activate the door control relay to open the door. Alternatively, a telephone can also originate a call to the Door Station, which would be automatically answered. The 8028 Doorphone is configured via a web interface.

2. General Test Approach and Test Results

To verify interoperability of the 8028 Doorphone with Communication Manager and Session Manager, calls were made from the doorphone to another specified telephone. The called telephone would ring and answer the call. Upon answering the call, a two-way audio path was established between the telephone and the Door Station. The called party would then be able to press a digit on the telephone keypad to open the door. In addition, incoming calls to the doorphone were also verified.

2.1. Interoperability Compliance Testing

Interoperability compliance testing covered the following features and functionality:

- Successful registration of the 8028 SIP Doorphone with Session Manager.
- Press call button at Door Station to ring specified telephone, answer the call, and establish a two-way audio path. Caller ID on the telephone was also verified.
- Called telephone can press a DTMF digit to open the door.
- Incoming calls to the 8028 Doorphone.
- G.711 codec support.
- Proper system recovery after the 8028 Doorphone loses power.

2.2. Test Results

All test cases passed and the 8028 SIP Doorphone successfully registered with Session Manager. Calls and delivery of DTMF tones to the doorphone were successful.

2.3. Support

For technical support on the 8028 SIP Doorphone, contact Algo Technical Support by phone, through their website, or email.

Phone:	(877) 884-2546 (Canada & US only)
Web:	http://www.algosolutions.com/support/support.html
Email:	support@algosolutions.com

3. Reference Configuration

Figure 1 illustrates a sample configuration with an Avaya SIP-based network that includes Session Manager, Communication Manager running on an Avaya S8800 Server with a G650 Media Gateway, and the Algo 8028 SIP Doorphone. Communication Manager was configured as an Evolution Server and the 8028 SIP Doorphone registered with Session Manager.

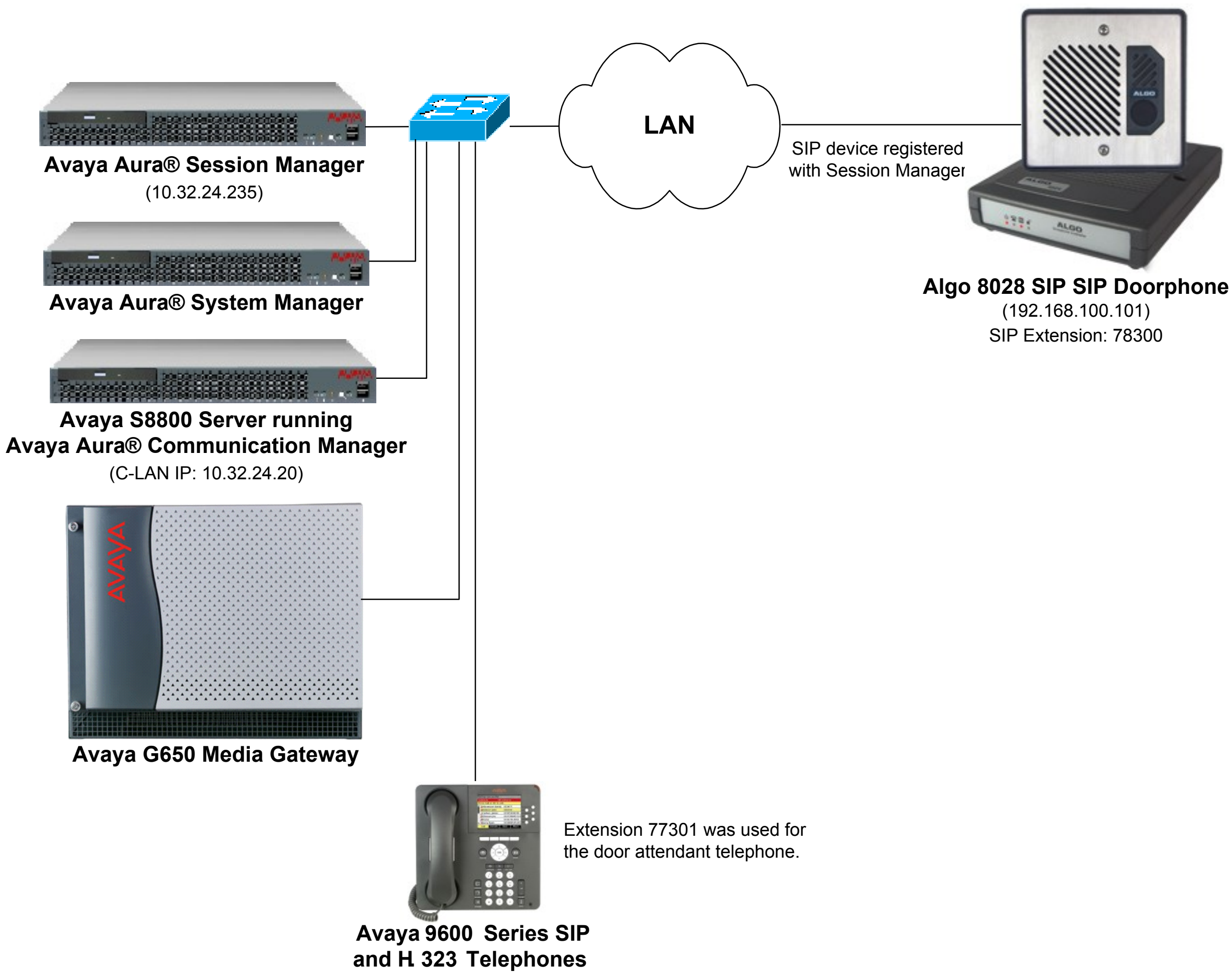


Figure 1: Avaya SIP Network with Algo 8028 SIP Doorphone

4. Equipment and Software Validated

The following equipment and software were used for the sample configuration provided:

Hardware Component	Version
Avaya S8800 Server and G650 Media Gateway	Avaya Aura® Communication Manager 6.0.1 SP 3 (R016x.00.1.510.1 w/Patch 19009)
Avaya Aura® Session Manager	6.1 (6.1.2.0-612004)
Avaya Aura® System Manager	6.1 (6.1.0.07345-6.1.5.106)
Avaya 9600 Series IP Telephones	3.1 (H.323) 2.6.4 (SIP)
Algo 8028 SIP Doorphone	1.4

5. Configure Avaya Aura® Communication Manager

This section describes the steps for configuring the 8028 SIP Doorphone as an Off-PBX Station (OPS) and configuring a SIP trunk between the Communication Manager and Session Manager. **Section 5.2** covers the station configuration for the 8028 SIP Doorphone. Use the System Access Terminal (SAT) to configure Communication Manager and log in with the appropriate credentials.

Note: If Communication Manager is already configured with a SIP trunk to Session Manager, skip **Section 5.1** and go directly to **Section 5.2** to configure the station for the 8028 SIP Doorphone.

5.1. Configure SIP Trunk

In the **IP Node Names** form, assign an IP address and host name for the S8800 Server processor, the C-LAN board in the G650 Media Gateway, and Session Manager. The host names will be used throughout the other configuration screens of Communication Manager.

```
change node-names ip                                           Page   1 of   2
                                  IP NODE NAMES
    Name              IP Address
Gateway001          10.32.24.1
ModMsg              192.50.10.45
clancrm             10.32.24.20
default             0.0.0.0
devcon-asm          10.32.24.235
medprocrm           10.32.24.21
procr               10.32.24.10
procr6              ::
 ( 8 of 8   administered node-names were displayed )
Use 'list node-names' command to see all the administered node-names
Use 'change node-names ip xxx' to change a node-name 'xxx' or add a node-name
```

In the **IP Network Region** form, the **Authoritative Domain** field is configured to match the domain name configured on Session Manager. In this configuration, the domain name is *avaya.com*. By default, **IP-IP Direct Audio** (shuffling) is enabled to allow audio traffic to be sent directly between IP endpoints without using media resources in the Avaya G650 Media Gateway. The **IP Network Region** form also specifies the **IP Codec Set** to be used for calls routed over the SIP trunk to Session Manager. This codec set is used when its corresponding network region (i.e., IP Network Region '1') is specified in the SIP signaling group.

```
change ip-network-region 1                                      Page   1 of  20
                               IP NETWORK REGION
  Region: 1
Location: 1       Authoritative Domain: avaya.com
    Name:
MEDIA PARAMETERS                Intra-region IP-IP Direct Audio: yes
      Codec Set: 1              Inter-region IP-IP Direct Audio: yes
   UDP Port Min: 2048                      IP Audio Hairpinning? y
   UDP Port Max: 65535
DIFFSERV/TOS PARAMETERS
 Call Control PHB Value: 34
        Audio PHB Value: 46
        Video PHB Value: 26
802.1P/Q PARAMETERS
 Call Control 802.1p Priority: 7
        Audio 802.1p Priority: 6
        Video 802.1p Priority: 5      AUDIO RESOURCE RESERVATION PARAMETERS
H.323 IP ENDPOINTS                                     RSVP Enabled? n
  H.323 Link Bounce Recovery? y
 Idle Traffic Interval (sec): 20
   Keep-Alive Interval (sec): 5
            Keep-Alive Count: 5
```

In the **IP Codec Set** form, select the audio codec type supported for calls routed over the SIP trunk to the 8028 SIP Doorphone. The form is accessed via the **change ip-codec-set 1** command. Note that IP codec set '1' was specified in IP Network Region '1' shown above. The default settings of the **IP Codec Set** form are shown below. The 8028 SIP Doorphone supports G.711.

```
change ip-codec-set 1                                           Page   1 of   2

                          IP Codec Set

    Codec Set: 1

    Audio        Silence      Frames   Packet
    Codec        Suppression  Per Pkt  Size(ms)
 1: G.711MU           n         2        20
 2:
 3:
 4:
 5:
 6:
 7:
```

Prior to configuring a SIP trunk group for communication with Session Manager, a SIP signaling group must be configured. Configure the signaling group form as follows:

- Set the **Group Type** field to *sip*.
- Set the **IMS Enabled** field to *n*.
- The **Transport Method** field was set to *tcp*. In a production network, TLS transport may also be used.
- Specify the C-LAN board and the Session Manager as the two ends of the signaling group in the **Near-end Node Name** field and the **Far-end Node Name** field, respectively. These field values are taken from the **IP Node Names** form.
- Ensure that the TCP port value of *5060* is configured in the **Near-end Listen Port** and the **Far-end Listen Port** fields.
- The preferred codec for the call will be selected from the IP codec set assigned to the IP network region specified in the **Far-end Network Region** field.
- Enter the domain name of Session Manager in the **Far-end Domain** field. In this configuration, the domain name is *avaya.com*.
- The **Direct IP-IP Audio Connections** field was enabled on this form.
- The **DTMF over IP** field should be set to the default value of *rtp-payload*. Communication Manager supports DTMF transmission using RFC 2833. The default values for the other fields may be used.

```
add signaling-group 50                                      Page   1 of   1
                                SIGNALING GROUP

 Group Number: 50            Group Type: sip
  IMS Enabled? n       Transport Method: tcp
        Q-SIP? n                                            SIP Enabled LSP? n
     IP Video? n                                  Enforce SIPS URI for SRTP? y
  Peer Detection Enabled? y  Peer Server: SM

   Near-end Node Name: clancrm              Far-end Node Name: devcon-asm
 Near-end Listen Port: 5060               Far-end Listen Port: 5060
                                       Far-end Network Region: 1
                                   Far-end Secondary Node Name:
Far-end Domain: avaya.com
                                             Bypass If IP Threshold Exceeded? n
Incoming Dialog Loopbacks: eliminate                  RFC 3389 Comfort Noise? n
         DTMF over IP: rtp-payload            Direct IP-IP Audio Connections? y
Session Establishment Timer(min): 3                     IP Audio Hairpinning? n
         Enable Layer 3 Test? n                    Initial IP-IP Direct Media? n
H.323 Station Outgoing Direct Media? n             Alternate Route Timer(sec): 6
```

Configure the **Trunk Group** form as shown below. This trunk group is used for calls to the SIP Phones. Set the **Group Type** field to *sip*, set the **Service Type** field to *tie*, specify the signaling group associated with this trunk group in the **Signaling Group** field, and specify the **Number of Members** supported by this SIP trunk group. Configure the other fields in bold and accept the default values for the remaining fields.

```
add trunk-group 50                                          Page   1 of  21
                                TRUNK GROUP

Group Number: 50                  Group Type: sip           CDR Reports: y
  Group Name: To devcon-asm              COR: 1        TN: 1        TAC: 1050
   Direction: two-way        Outgoing Display? n
 Dial Access? n                                    Night Service:
Queue Length: 0
Service Type: tie                   Auth Code? n
                                             Member Assignment Method: auto
                                                      Signaling Group: 50
                                                    Number of Members: 10
```

5.2. Configure Station

Use the **add station** command to add station for the 8028 SIP Doorphone. Use *9640SIP* for the **Station Type**. The **Name** field is optional. Use the default values for the other fields. The SIP station can also be configured automatically by Session Manager as described in **Section 6.7**.

```
add station 78300                                            Page   1 of   6
                                     STATION

Extension: 78300                        Lock Messages? n               BCC: 0
     Type: 9640SIP                      Security Code:                  TN: 1
     Port: IP                         Coverage Path 1:                 COR: 1
     Name: 78300, Doorphone           Coverage Path 2:                 COS: 1
                                      Hunt-to Station:
STATION OPTIONS
                                          Time of Day Lock Table:
              Loss Group: 19
                                                Message Lamp Ext: 78300

        Display Language: english                 Button Modules: 0

          Survivable COR: internal
   Survivable Trunk Dest? y                         IP SoftPhone? n

                                                        IP Video? n
```

Use the **change off-pbx-telephone station-mapping** command to map the Communication Manager extension to the same extension on Session Manager. Enter the field values shown below. For the sample configuration, the **Trunk Selection** field is set to *aar* so that AAR call routing is used to route calls to Session Manager. AAR call routing configuration is not shown in these Application Notes. The **Config Set** value can reference a set that has the default settings.

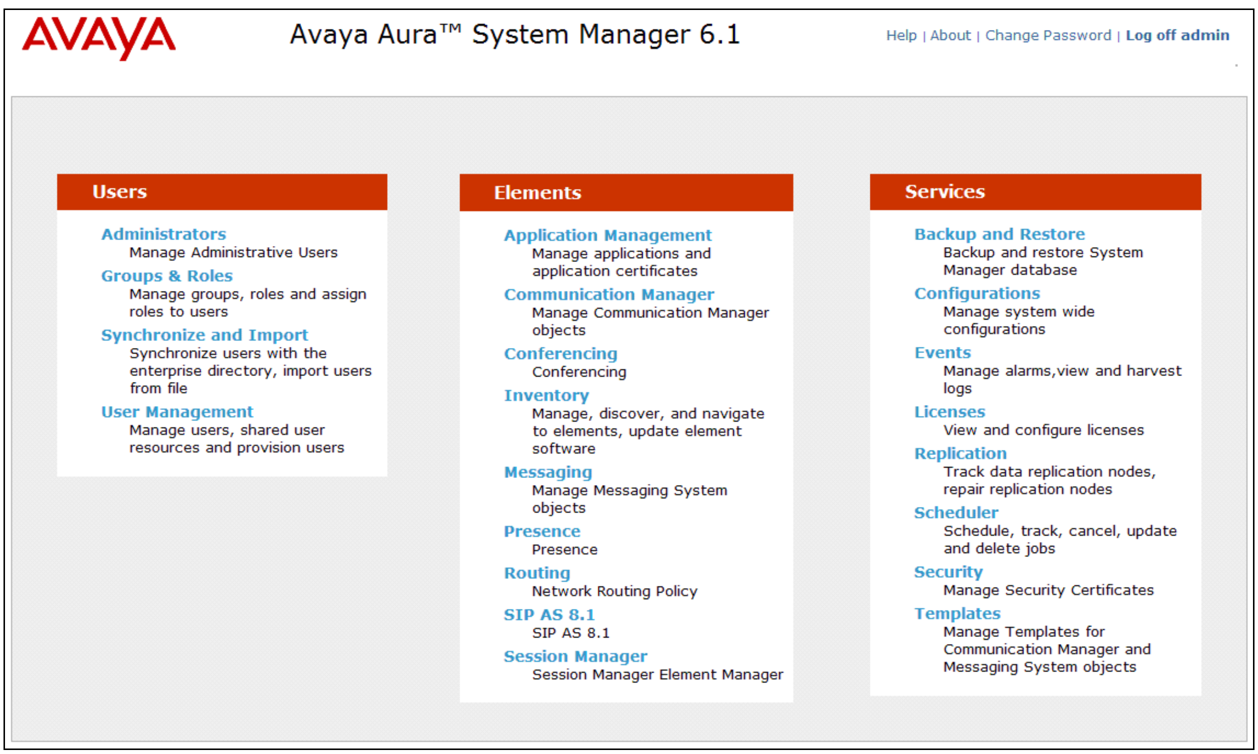
```
change off-pbx-telephone station-mapping 78300                  Page   1 of   3
                  STATIONS WITH OFF-PBX TELEPHONE INTEGRATION

 Station       Application Dial   CC  Phone Number    Trunk        Config  Dual
 Extension                 Prefix                     Selection    Set     Mode
 78300            OPS          -      78300            aar         1
```

6. Configure Avaya Aura® Session Manager

This section provides the procedures for configuring Session Manager. The procedures include adding the following items:

- SIP domain
- Logical/physical Locations that can be occupied by SIP Entities
- SIP Entities corresponding to Session Manager and Communication Manager
- Entity Links, which define the SIP trunk parameters used by Session Manager when routing calls to/from SIP Entities
- Define Communication Manager as Administrable Entity (i.e., Managed Element)
- Application Sequence
- Add SIP User for the 8028 SIP Doorphone
- Session Manager, corresponding to the Session Manager Server to be managed by System Manager

Configuration of Session Manager is accomplished by accessing the browser-based GUI of System Manager using the URL "https://<*ip-address*>/SMGR", where <*ip-address*> is the IP address of System Manager. Log in with the appropriate credentials. The initial screen is displayed as shown below. The configuration in this section will be performed under **Routing** and **Session Manager** listed within the **Elements** box.



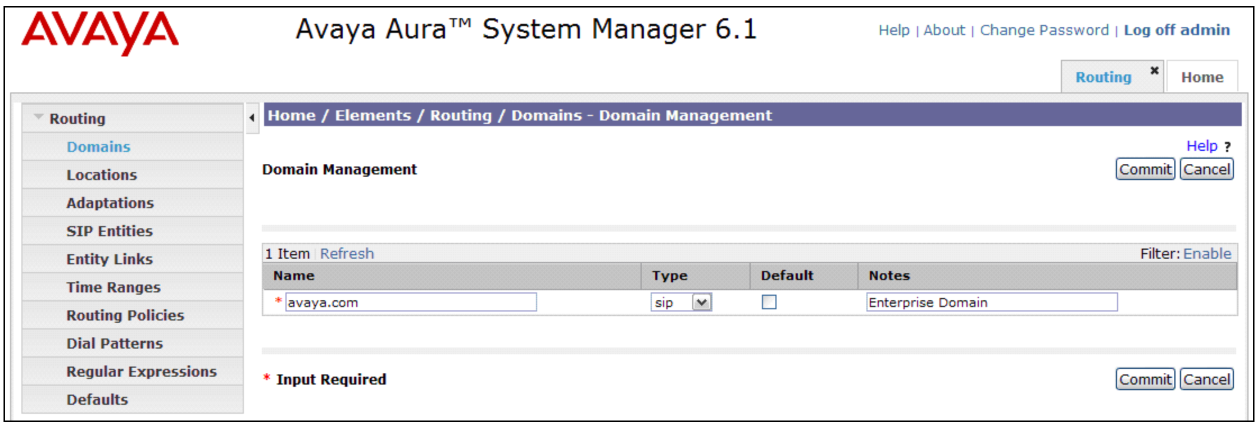
6.1. Specify SIP Domain

Add the SIP domain for which the communications infrastructure will be authoritative. Do this by selecting **Domains** on the left and clicking the **New** button (not shown) on the right. The following screen will then be shown. Fill in the following:

- **Name:** The authoritative domain name (e.g., *avaya.com*)
- **Notes:** Descriptive text (optional).

Click **Commit**.

Since the sample configuration does not deal with any other domains, no additional domains need to be added.



6.2. Add Locations

Locations can be used to identify logical and/or physical locations where SIP Entities reside for purposes of bandwidth management. To add a location, select **Locations** on the left and click on the **New** button (not shown) on the right. The following screen will then be shown. Fill in the following:

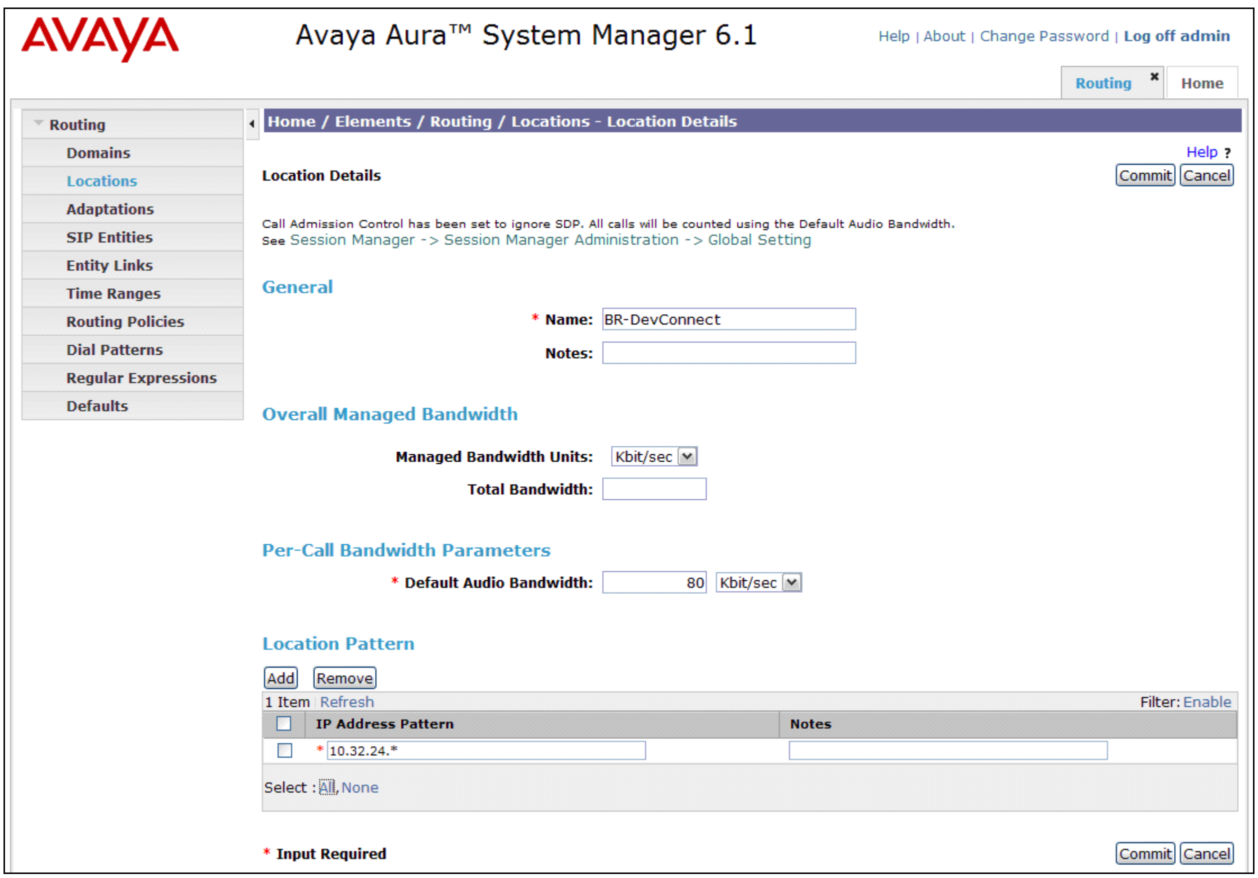
Under *General*:

- **Name:** A descriptive name.
- **Notes:** Descriptive text (optional).

Under *Location Pattern*:

- **IP Address Pattern:** A pattern used to logically identify the location.
- **Notes:** Descriptive text (optional).

The screen below shows addition of the *BR-DevConnect* location, which includes the Communication Manager and Session Manager. Click **Commit** to save the Location definition.



6.3. Add SIP Entities

In the sample configuration, a SIP Entity is added for Session Manager and the C-LAN in the G650 Media Gateway.

6.3.1. Session Manager

A SIP Entity must be added for Session Manager. To add a SIP Entity, select **SIP Entities** on the left and click on the **New** button (not shown) on the right. The following screen is displayed. Fill in the following:

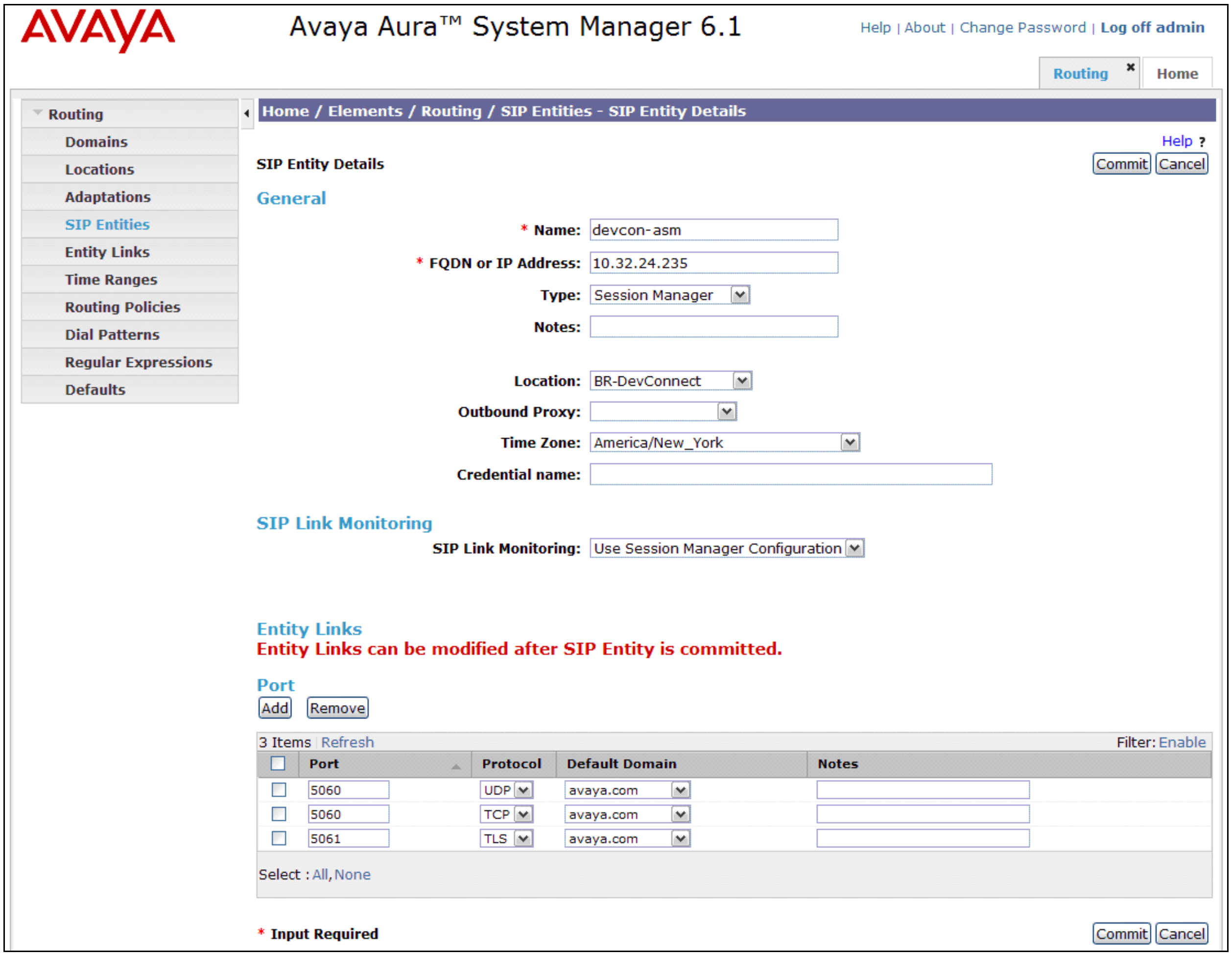
Under *General*:

- **Name:** A descriptive name.
- **FQDN or IP Address:** IP address of the signaling interface on Session Manager.
- **Type:** Select *Session Manager*.
- **Location:** Select the location defined previously.
- **Time Zone:** Time zone for this location.

Under *Port*, click **Add**, and then edit the fields in the resulting new row as shown below:

- **Port:** Port number on which the system listens for SIP requests.
- **Protocol:** Transport protocol to be used to send SIP requests.
- **Default Domain** The domain used for the enterprise (e.g., *avaya.com*).

Defaults can be used for the remaining fields. Click **Commit** to save each SIP Entity definition.


AVAYA
Avaya Aura™ System Manager 6.1
Help | About | Change Password | Log off admin
Routing
Home
Routing
Domains
Locations
Adaptations
SIP Entities
Entity Links
Time Ranges
Routing Policies
Dial Patterns
Regular Expressions
Defaults
Home / Elements / Routing / SIP Entities - SIP Entity Details
Help ?
SIP Entity Details
Commit Cancel
General
* Name: devcon-asm
* FQDN or IP Address: 10.32.24.235
Type: Session Manager
Notes:
Location: BR-DevConnect
Outbound Proxy:
Time Zone: America/New_York
Credential name:
SIP Link Monitoring
SIP Link Monitoring: Use Session Manager Configuration
Entity Links
Entity Links can be modified after SIP Entity is committed.
Port
Add Remove
3 Items Refresh
Filter: Enable
Port Protocol Default Domain Notes
5060 UDP avaya.com
5060 TCP avaya.com
5061 TLS avaya.com
Select : All, None
* Input Required
Commit Cancel

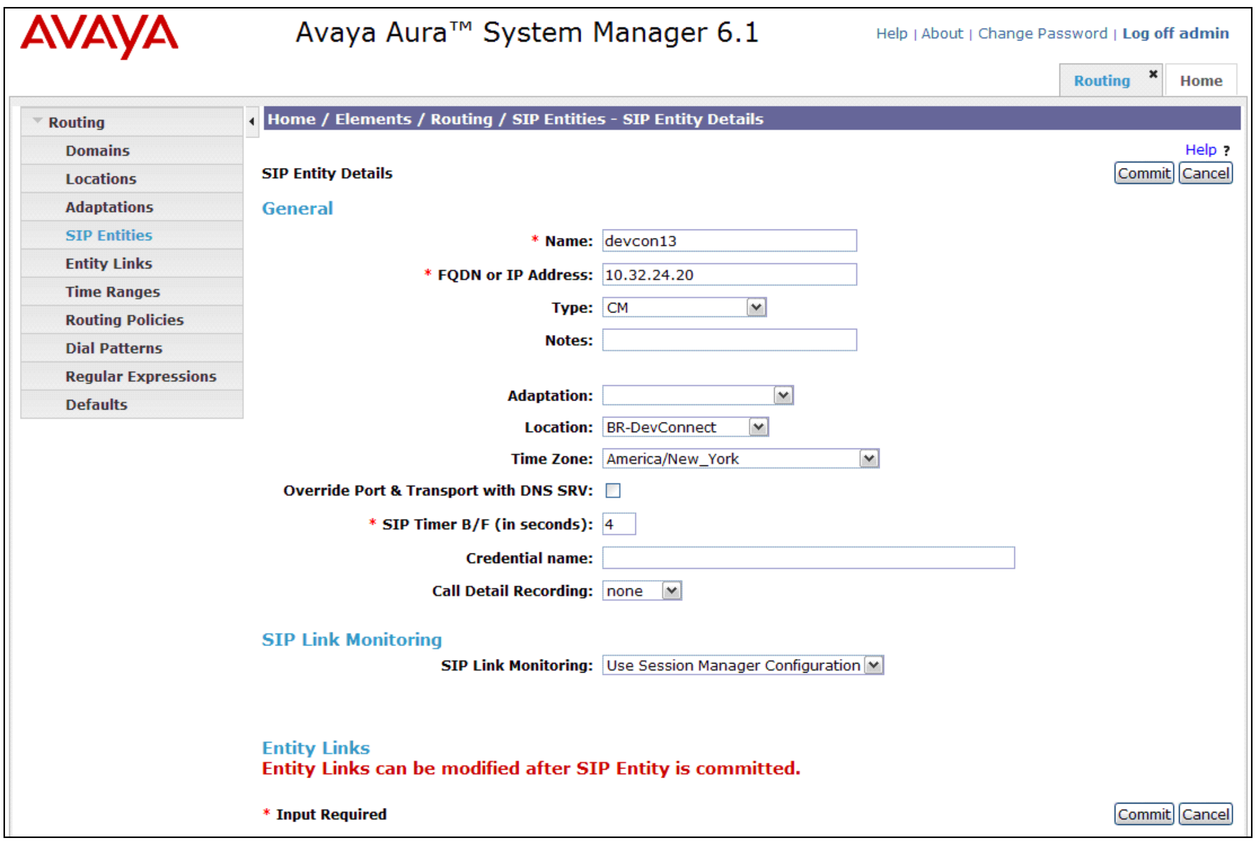
6.3.2. Communication Manager

A SIP Entity must be added for the Communication Manager. To add a SIP Entity, select **SIP Entities** on the left and click on the **New** button (not shown) on the right. The following screen is displayed. Fill in the following:

Under *General*:

- **Name:** A descriptive name.
- **FQDN or IP Address:** IP address of the signaling interface (e.g., C-LAN board) on the telephony system.
- **Type:** Select *CM*.
- **Location:** Select the location defined previously.
- **Time Zone:** Time zone for this location.

Defaults may be used for the remaining fields. Click **Commit** to save the SIP Entity definition.

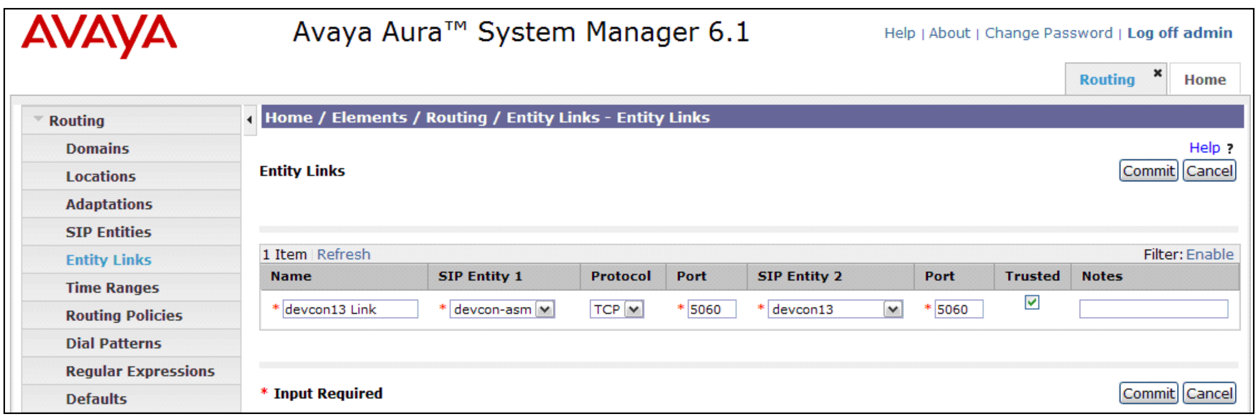


6.4. Add Entity Link

The SIP trunk from Session Manager to Communication Manager is described by an Entity link. To add an Entity Link, select **Entity Links** on the left and click on the **New** button (not shown) on the right. Fill in the following fields in the new row that is displayed:

- **Name:** A descriptive name (e.g., *devcon13 Link*).
- **SIP Entity 1:** Select the Session Manager.
- **Protocol:** Select the appropriate protocol.
- **Port:** Port number to which the other system sends SIP requests.
- **SIP Entity 2:** Select the name of Communication Manager.
- **Port:** Port number on which the other system receives SIP requests.
- **Trusted:** Check this box. *Note: If this box is not checked, calls from the associated SIP Entity specified in* ***Section 6.3.2*** *will be denied.*

Click **Commit** to save the Entity Link definition.



6.5. Define Communication Manager as Managed Element

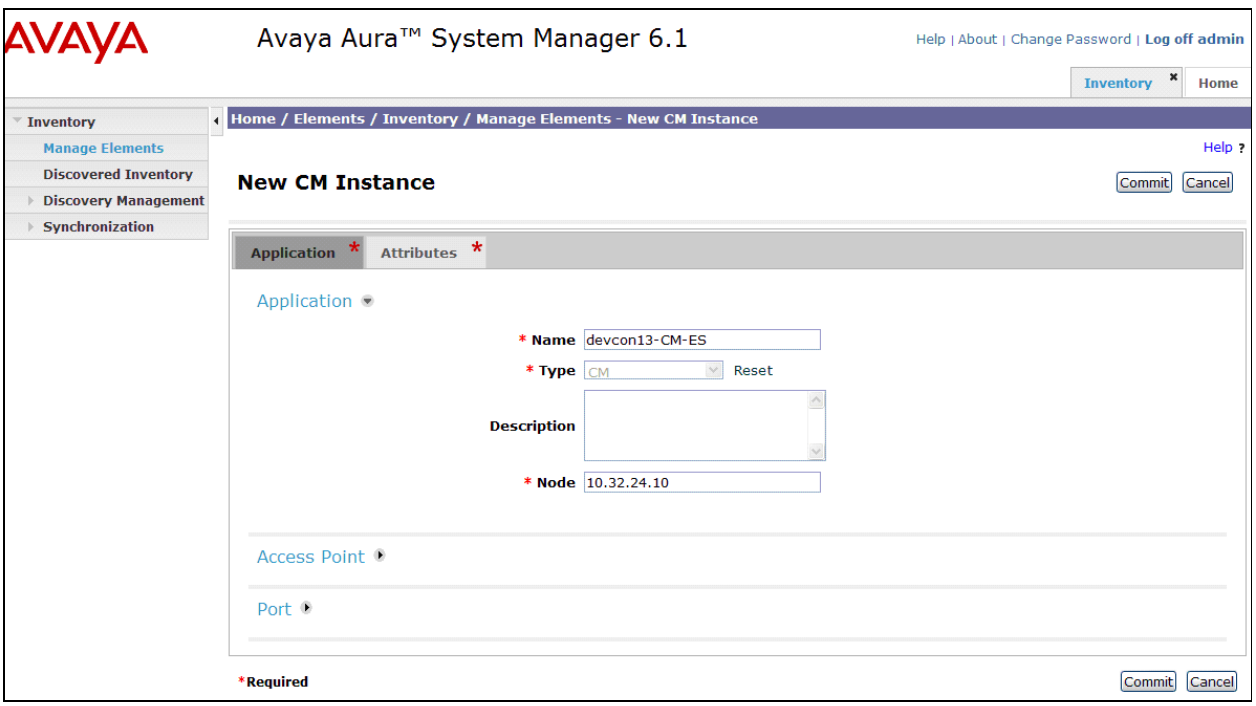
Before adding SIP users, Communication Manager must be added to System Manager as a managed element. This action allows System Manager to access Communication Manager over its administration interface. Using this administration interface, System Manager will notify Communication Manager when new SIP users are added.

To define Communication Manager as a managed element, select **Elements→Inventory→Manage Elements** on the left and click on the **New** button (not shown) on the right. In the **New Entities Instance** screen (not shown), select *CM* in the **Type** field, then click **Commit**.

In the **New CM Instance** screen, fill in the following fields as follows:

In the *Application* tab:

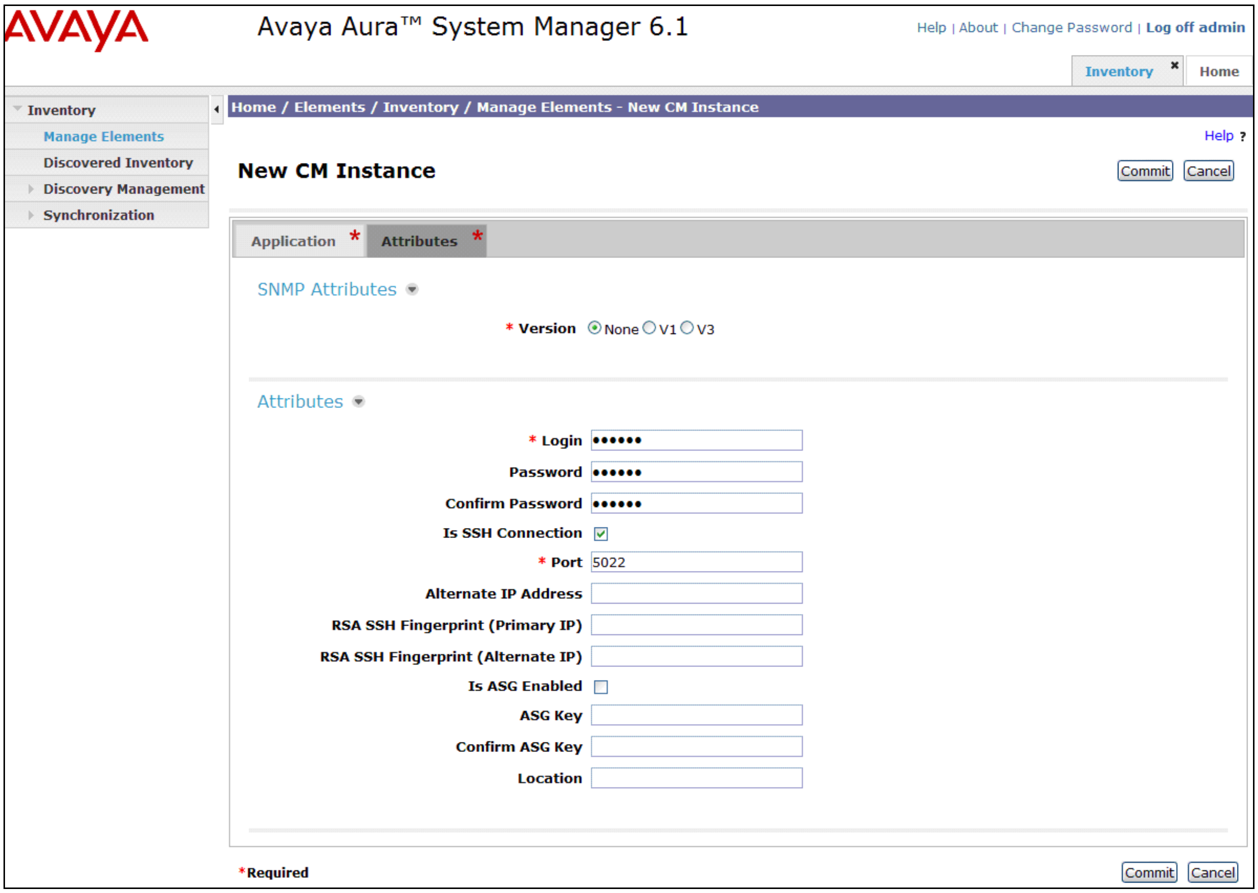
- **Name:** Enter an identifier for Communication Manager.
- **Type:** Select *CM* from the drop-down field.
- **Node:** Enter the IP address of the administration interface for Communication Manager.



In the *Attributes* tab:

- **Login / Password:** Enter the login and password used for administration access.
- **Is SSH Connection:** Enable SSH access.
- **Port:** Enter the port number for SSH administration access (5022).

Defaults can be used for the remaining fields. Click **Commit** to save the settings.

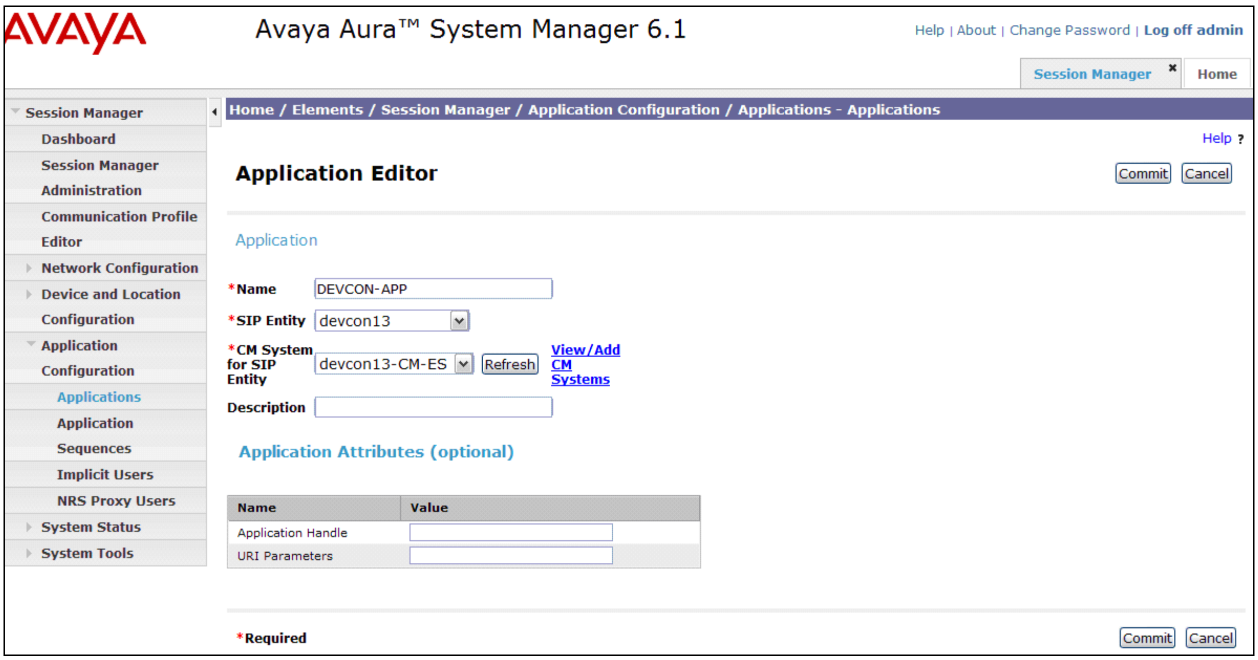


6.6. Add Application Sequence

To define an application for Communication Manager, navigate to **Elements → Session Manager → Application Configuration → Applications** on the left and select **New** button (not shown) on the right. Fill in the following fields:

- **Name:** Enter name for application.
- **SIP Entity:** Select the Communication Manager SIP entity.
- **CM System for SIP Entity** Select the Communication Manager managed element.

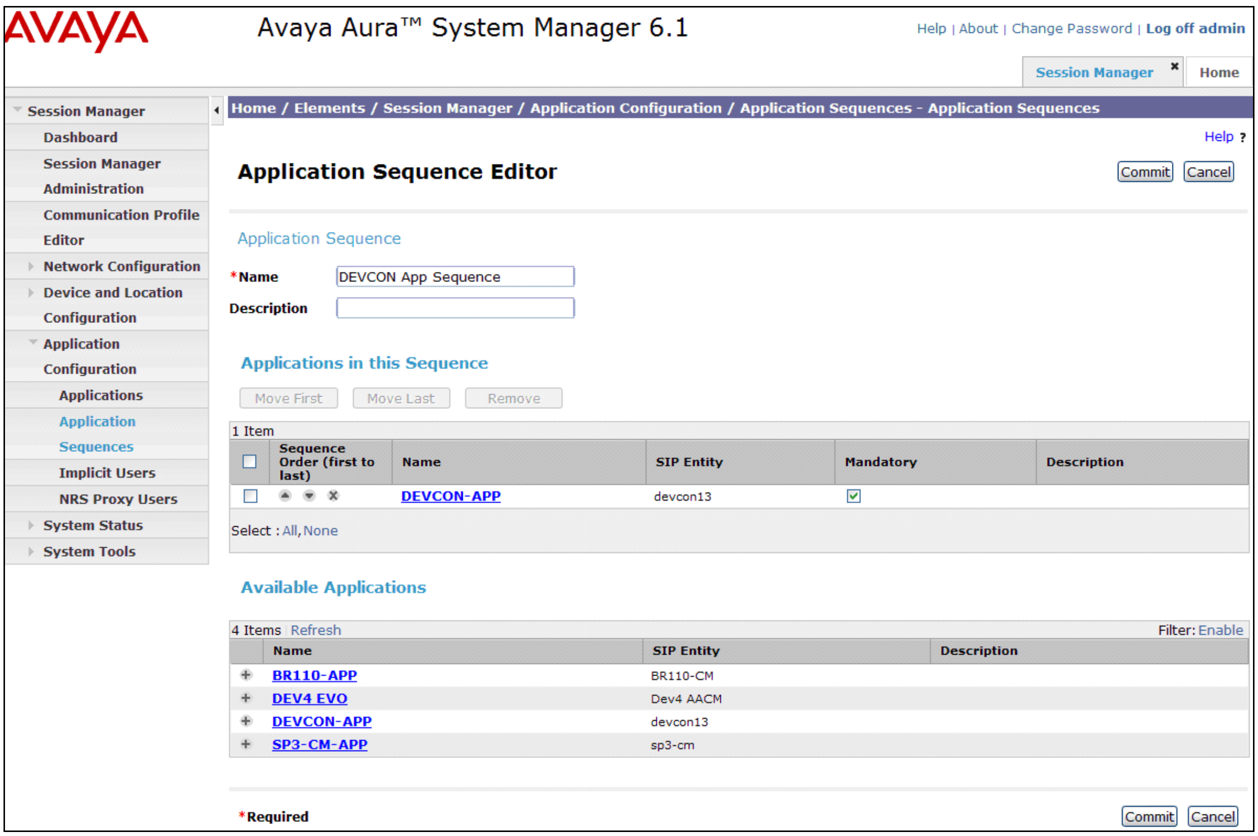
Click **Commit** to save the Application definition.



Next, navigate to **Elements → Session Manager → Application Configuration → Application Sequences** to define the Application Sequence for Communication Manager as shown below. Provide a **Name** (e.g., *DEVCON App Sequence*) for the Application Sequence and under **Available Applications**, click on the plus (+) sign by *DEVCON-APP* to add it under the **Application in this sequence** section.

Verify a new entry is added to the **Applications in this Sequence** table and the **Mandatory** column is ☑ as shown below.

Note: The Application Sequence defined for Communication Manager Evolution Server can only contain a single Application.



6.7. Add SIP User

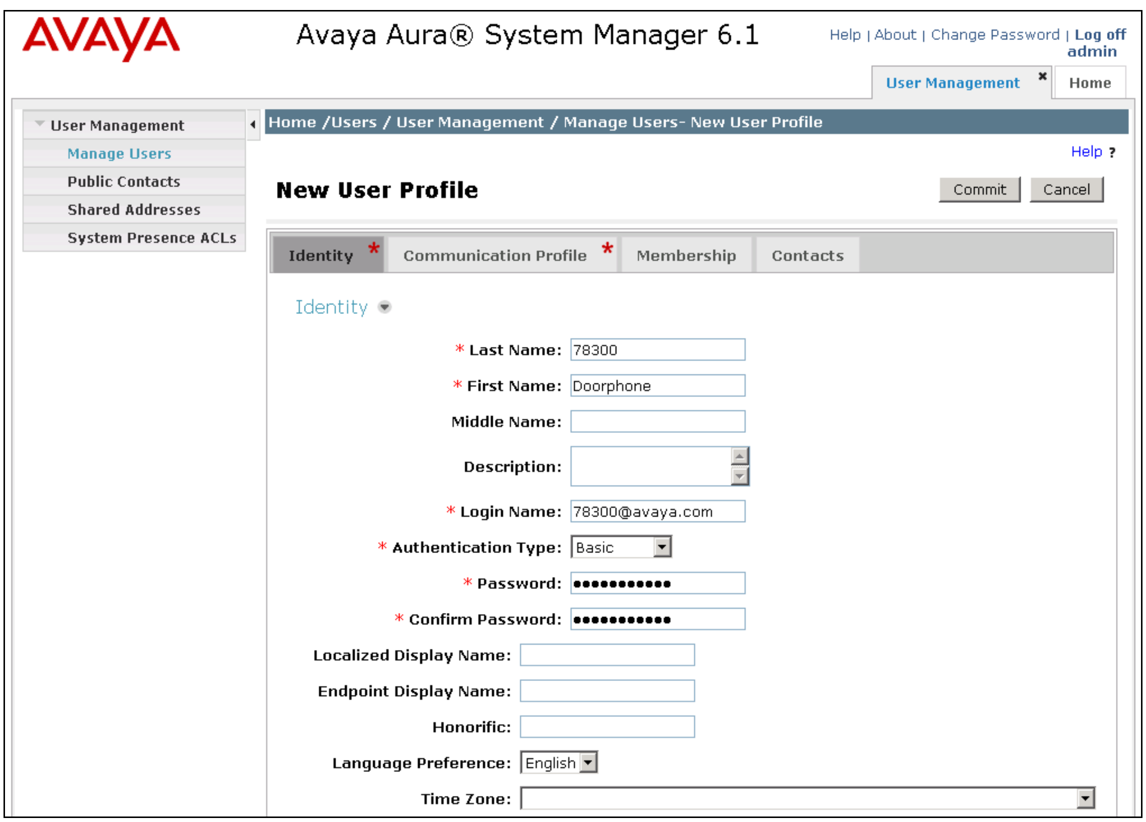
Add a SIP user for the 8028 SIP Doorphone**.** The following configuration will automatically create the SIP station on Communication Manager Evolution Server.

To add new SIP users, navigate to **Users → User Management → Manage Users** from the left and select **New** button (not shown) on the right.

Enter values for the following required attributes for a new SIP user in the **Identity** tab of the new user form.

- **Last Name:** Enter the last name of the user.
- **First Name:** Enter the first name of the user.
- **Login Name:** Enter <*extension*>@<*sip domain*> of the user (e.g., 78300@avaya.com).
- **Authentication Type:** Select *Basic***.**
- **Password:** Enter the password which will be used to log into System Manager
- **Confirm Password:** Re-enter the password from above.

The screen below shows the information when adding a new SIP user to the sample configuration.



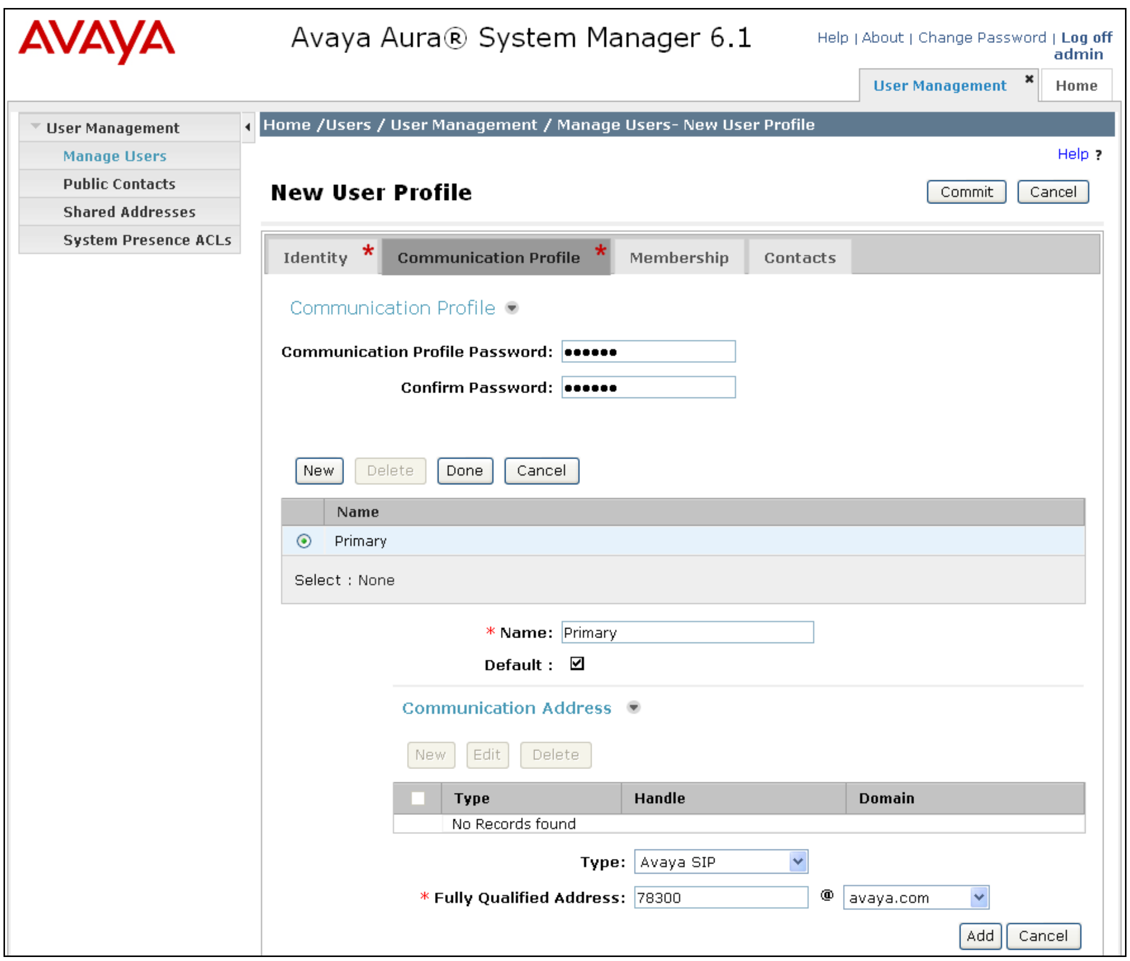
Enter values for the following required attributes for a new SIP user in the **Communication Profile tab** of the new user form.

- **Communication Profile Password:** Enter the password which will be used by the 8028 SIP Doorphone to register with Session Manager.
- **Confirm Password:** Re-enter the password from above.

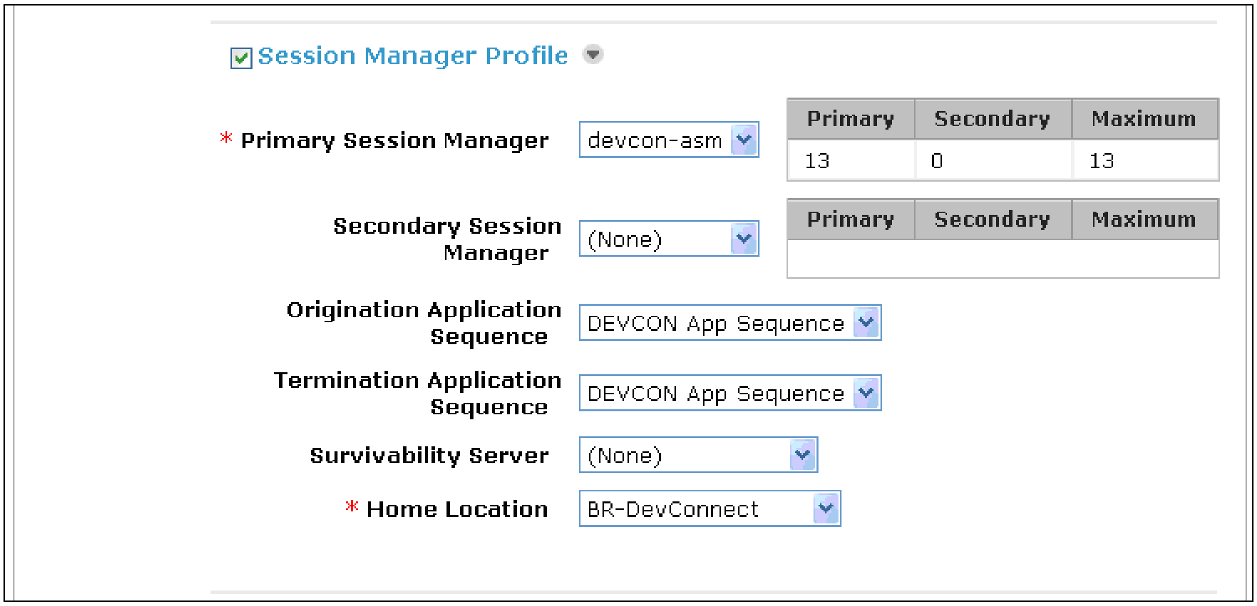
Scroll down to the **Communication Address** section and select **New** to define a **Communication Address** for the new SIP user. Enter values for the following required fields:

- **Type:** Select *Avaya SIP*.
- **Fully Qualified Address:** Enter extension number and select SIP domain.

The screen below shows the information when adding a new SIP user to the sample configuration. Click **Add**.



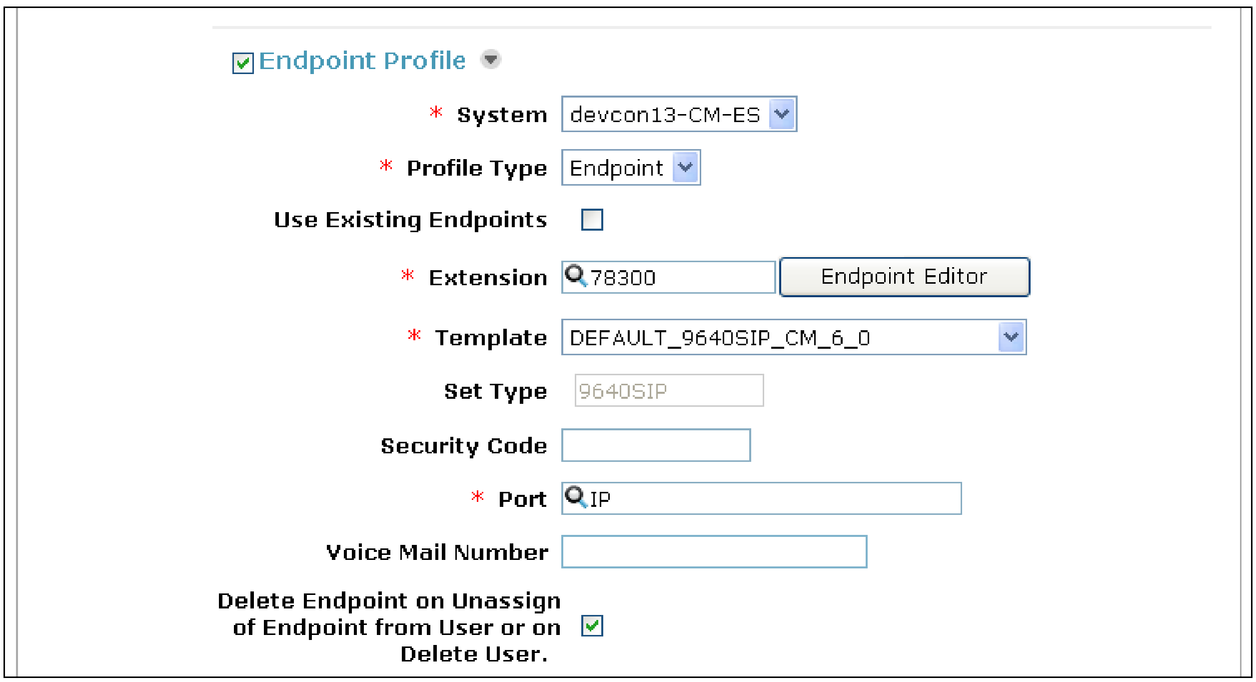
In the *Session Manager Profile* section, specify the Session Manager entity from **Section 6.3.1** for **Primary Session Manager** and assign the **Application Sequence** defined in **Section 6.6** to the new SIP user as part of defining the **SIP Communication Profile**. The **Application Sequence** can be used for both the originating and terminating sequence. Set the **Home Location** field to the **Location** configured in **Section 6.2**.



In the **Endpoint Profile** section, fill in the following fields:

- **System:** Select the managed element corresponding to Communication Manager.
- **Profile Type** Select *Endpoint*.
- **Use Existing Stations:** If field is not selected, the station will automatically be added in Communication Manager.
- **Extension:** Enter extension number of SIP user.
- **Template:** Select template for type of SIP phone.
- **Port:** Enter *IP*.
- **Delete Endpoint on Unassign of Endpoint:** Enable field to automatically delete station when **Endpoint Profile** is un-assigned from user.

Note: To specify a coverage path for voicemail, click on the **Endpoint Editor** button.



6.8. Add Session Manager

To complete the configuration, adding the Session Manager will provide the linkage between System Manager and Session Manager. Expand the **Session Manager** menu on the left and select **Session Manager Administration**. Click **Add** (not shown), and fill in the fields as described below and shown in the following screen:

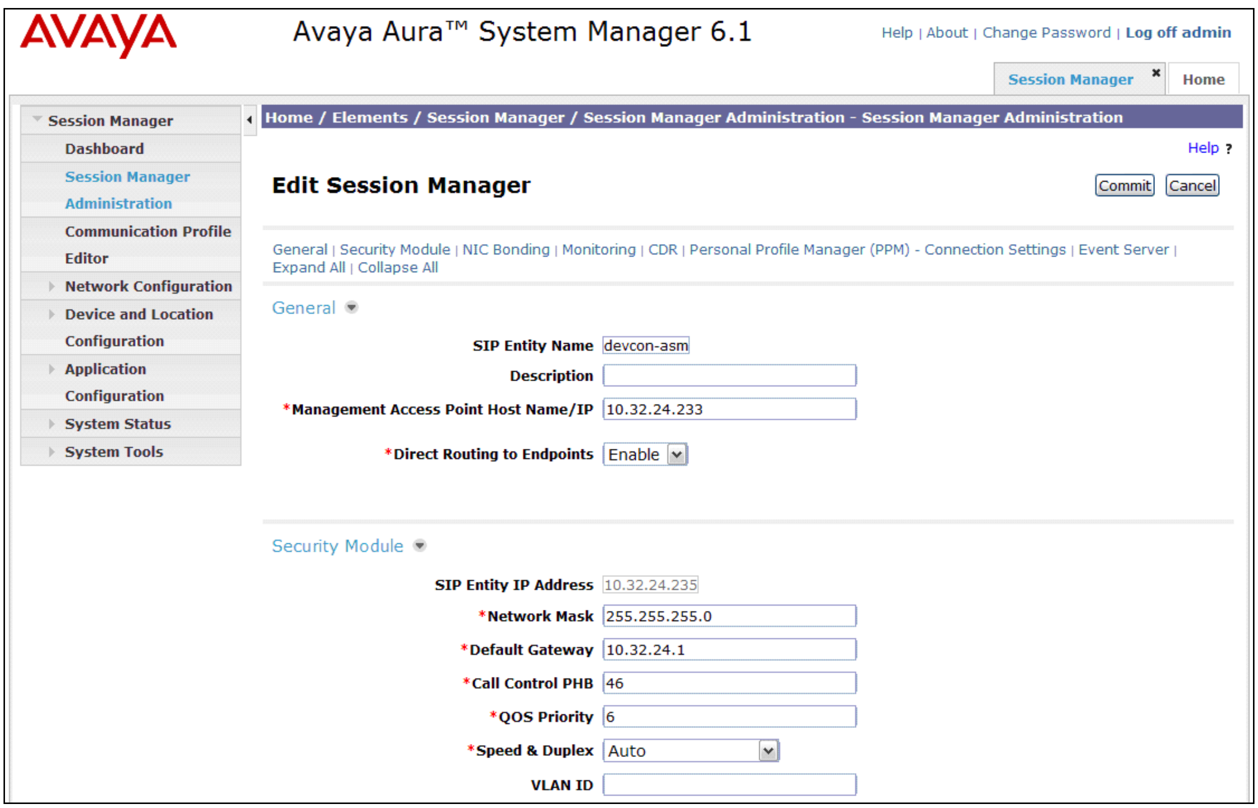
Under *Identity*:

- **SIP Entity Name:** Select the name of the SIP Entity added for Session Manager
- **Description:** Descriptive comment (optional)
- **Management Access Point Host Name/IP:** Enter the IP address of the Session Manager management interface.

Under *Security Module*:

- **Network Mask:** Enter the network mask corresponding to the IP address of Session Manager
- **Default Gateway**: Enter the IP address of the default gateway for Session Manager

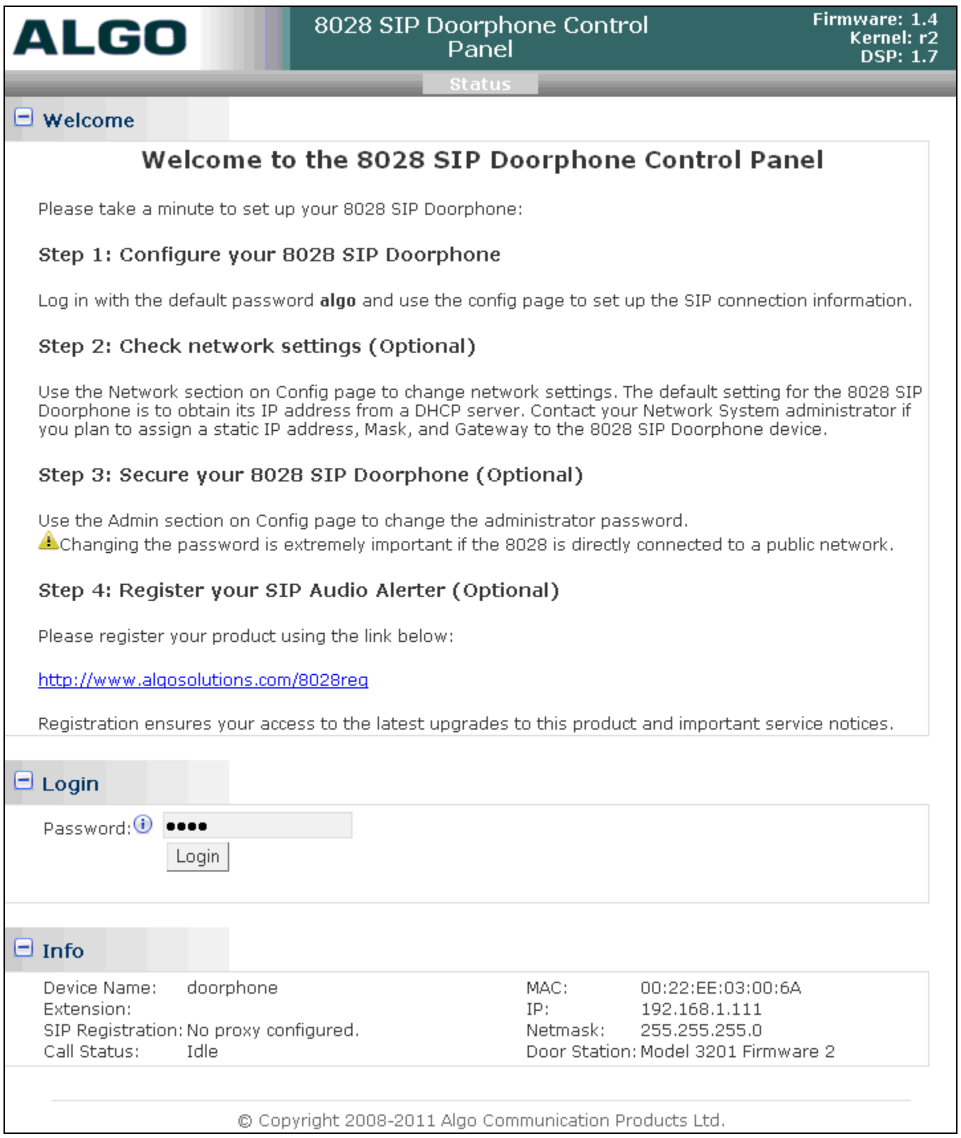
Use default values for the remaining fields. Click **Commit** to add this Session Manager.



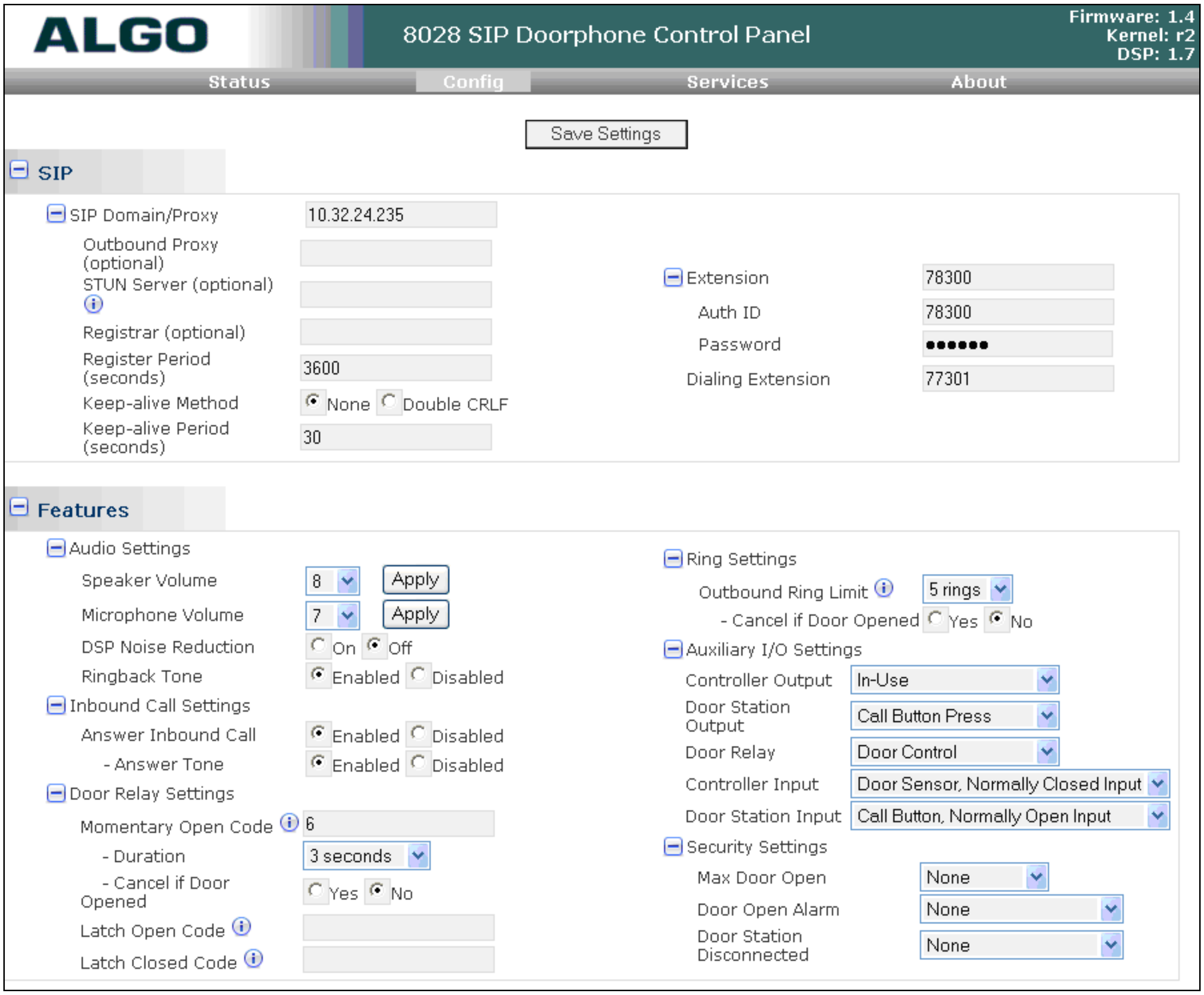
7. Configure Algo 8028 SIP Doorphone

The configuration of the 8028 SIP Doorphone was performed by using a web interface tool, which is accessed by entering the 8028 IP address into a browser. The 8028 supports DHCP as the default method to assign an IP address to the unit. Alternatively, if DHCP is not available, a pre-assigned IP address may be used to access the unit. The default IP address is 192.168.1.111. A valid password will be required. Refer to **[3]** for more information, including installation.

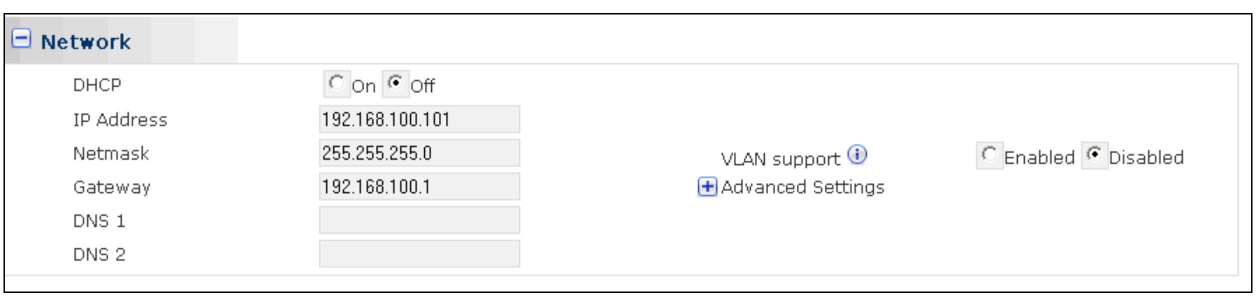
From an internet browser, enter http://<*ip-addr*> in the URL field, where <*ip-addr*> is the IP address of the 8028 SIP Doorphone. The screen shown below is displayed. Log in with the appropriate credentials.



In the **Config** tab, set the **SIP Domain/Proxy** field to the Session Manager IP address. This should be the IP address to which the 8028 SIP Doorphone will send SIP messages. Specify the **Extension** and **Auth ID**, along with the **Password**, configured in **Section 6.7**. The **Dialing Extension** (e.g., *77301*) should specify the telephone number that will be dialed when the call button on the Door Station is pressed.



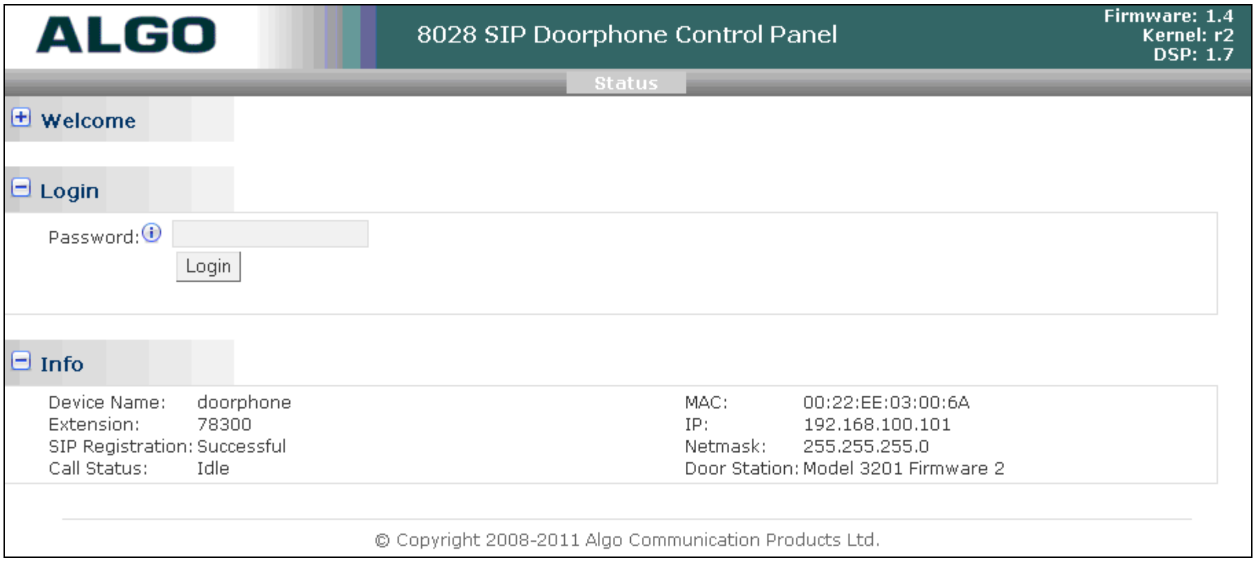
Finally, configure the 8028 with the appropriate IP network parameters. Use the configured IP address to access the 8028 instead of the default IP address. Click **Save Settings**.



8. Verification Steps

The following steps can be used to verify and/or troubleshoot installations in the field.

1. Verify that the 8028 SIP Doorphone has successfully registered with Session Manager. The **SIP Registration** field in the **Status** tab should indicate successful registration as shown below.



2. Verify that when the call button on the Door Station is pressed, the specified telephone on Communication Manager rings, and upon answering the call, two-way audio path is established.

3. Verify that the 8028 SIP Doorphone returns to the idle state when the call is terminated.

4. Verify that incoming calls to the 8028 SIP Doorphone are also successful.

9. Conclusion

These Application Notes describe the administration steps required to integrate the Algo 8028 SIP Doorphone with Avaya Aura® Communication Manager and Avaya Aura® Session Manager. The 8028 SIP Doorphone successfully registered with Session Manager and incoming and outgoing calls were successful. In addition, unlocking a door using DTMF tones was successful. All test cases passed.

10. Additional References

This section references documentation relevant to these Application Notes. The following Avaya product documentation is available at http://support.avaya.com.

[1] *Administering Avaya Aura™ Communication Manager*, June 2010, Release 6.0, Issue 6.0, Document Number 03-300509.
[2] *Administering Avaya Aura® Session Manager*, November 2010, Issue 1.1, Release 6.1, Document Number 03-603324.

The following Algo product documentation is available at http://www.algosolutions.com.

[3] *Algo 8028 SIP Doorphone Installation and User Guide*, Document Number 90-00054A.

Please e-mail any questions or comments pertaining to these Application Notes along with the full title name and filename, located in the lower right corner, directly to the Avaya DevConnect Program at devconnect@avaya.com.